MiRACALL *Miracall Technologies*

IP PBX MC-IP8008
User Manual

Please read this manual carefully before operation

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Chapter 1 Brief introduction & Function

MiraCALL04/08 is an embedded ippbx based on standard asterisk for SMEs, which is not only a PBX, but also as a voice mail Server, IVR server, conferencing server. With 4 or 8 analog interface which can be configured as FXS or FXO ports (made in factory), and 1Wan and 1Lan with router function. With excellent echo canceller function, it can meet most of the customers requirement.

1.1 System Features

* Based on Asterisk
* Configuration by Web
* Built-in SIP/IAX Server
* Static/DHCP/PPPoE network access
* Codec: G.711-Ulaw, G.711-Alaw, G.726, G.729, GSM, SPEEX
* SIP/IAX Trunk(use with VoIP Trunk operator)
* Zap Trunk(Use with PSTN)
* SIP/IAX Extensions(connect with IP Phone)
* Zap Extensions(connect with Analog Phone)
* Voice Mail ,and voicemail to email
* Flexible Dial Plan
* Call Conference
* Auto-attendant
* Music On Hold
* Call Queue
* Call history list
* Support IP Phone with Key function
* FAX T.38
* Voicemail to email
* Other basic function:
    1. Three way conference
    2. Call Forward(and forward to external PSTN or mobile)
    3. Call Hold
    4. Call Transfer
    5. Call Waiting
    6. Caller ID

1.2 Specifications

1) Interface

* 4or 8 * Analog Ports (RJ11)
* 1 * USB Interface(for Wifi moudule, unavailable)
* 1 * SD MMC Interface (2GB,to install voicemail)
* 2 * Network Interface (RJ45,1 for WAN, 1 for LAN)
* 1 * Power port（DC 12V 2A）
* 1 * Re-set Button (beside PWR button,black)

2) Panel

Mark	Function	Status	Description
PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	On	System working
		Off	System disabled
WAN	WAN interface Status	Wink	Data exchanging
		Off	No Data exchanging
LAN	LAN Interface Status	Wink	Data exchanging
		Off	No Data exchanging
MMC	SD card Status	On	MMC enabled
		Off	MMC disabled
USB	USB WiFi module	On	WiFi enabled
		Off	WiFi disabled
1-8lamp	Analog Modules Status	Red	FXO port
		Green	FXS port
		Off	FXO or FXS no connection

3) Hardware
   * 32bit embedded RISC DSP
   * 256M Onboard Nand Flash
   * 2M Onboard Nor Flash
   * 64M Onboard SDRAM
   * 2G MMC/SD Storage
4) environmental requirements:
   * temperature: -10 °C -48 °C
   * Storage temperature: -30 °C -68 °C
   * humidity: 10-80% no dew
   * Power: 90~260VAC
5) Packing List
   * MC-IP4008,8008 1 Unit
   * Power Adapter 1 Unit
   * CD 1 pc
6) Packing Size:
   * Inner box MEAS: 280*198*70mm
   * G.W/unit: 1.3kg
   * Carton MEAS: 520*580*220mm
   * G.W: 15kg
   * PCS/CTN: 10PCS

1.3 Default configuration

1. Wan port IP address: 192.168.1.100:9999
2. Lan port IP address: 192.168.10.100:9999
3. Web GUI username: admin
4. Web GUI password: admin
5. Command line username: root
6. Command line password: myippbx

Chapter2 Basic Configuration

2.1. Log into the system

After connecting the ippbx to the local area network. Launch the web browser on a computer that is in this local area network. Enter the IP address for the system (Wan port IP address 192.168.1.100:9999, Lan port IP address 192.168.10.100:9999) . The start web page will appear like this:

Enter Username and password (default username is admin, password is admin), then click login. Once the login is success the home page will be display:

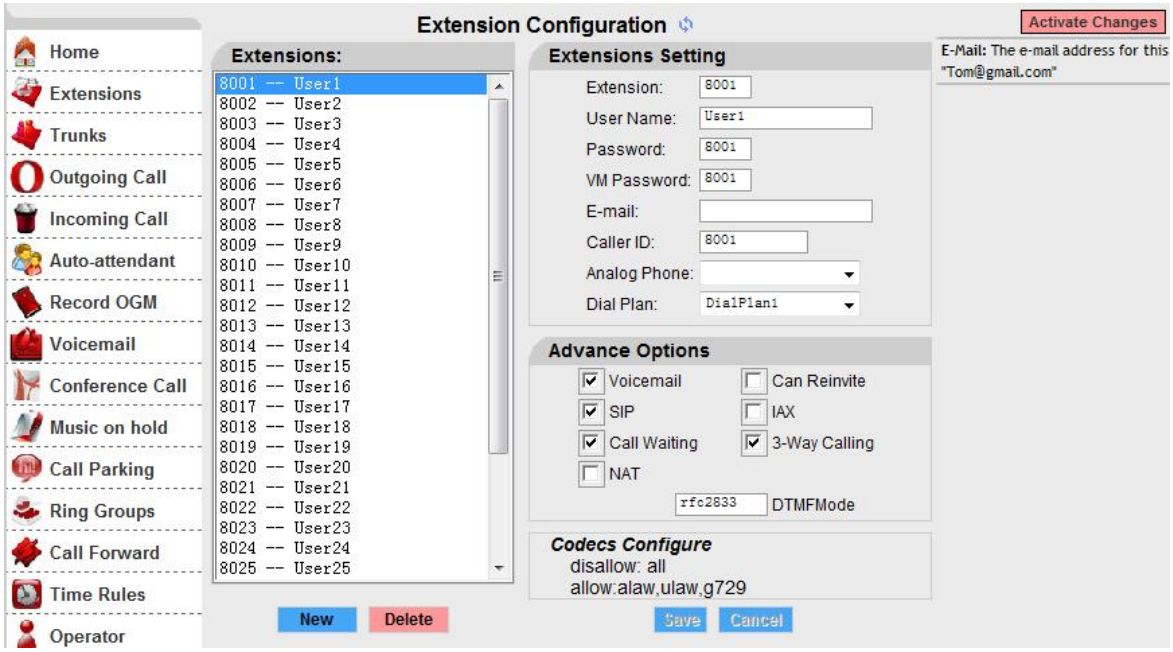
With the MiraCALL GUI, you can configure extensions, conference, voicemail, Dial Plan

The home page is used for login out, Reboot or return to Factory default status.

* Logout: To log out the MiraCALL GUI.
* Restart Asterisk Restarted the Asterisk system.
* Reboot: Reboot the IP-PBX system
* Factory Defaults: Restore all settings to de factory default.
* Activate change: Made the change active for the current configuration after you make a configuration change on some page.Everytime you should activate after change the command or setting.

2.2. Configure User Extensions

Click the Users tab and you will see the extensions setting, your created user in this page.



The following information comprises a user extension definition:

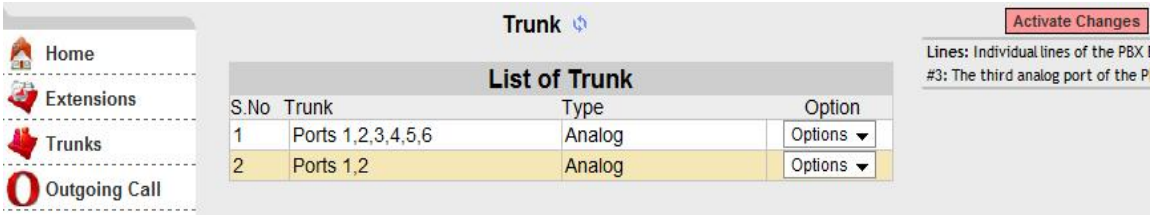
* Extension The extension number given to station user.
* Name The name for this extension, can be only.
* Password Password to register the extension, same as IP phone
* VM Password Password used to access specified voicemail extension.
* E-mail Address Voice mails received by this extension can be sent as audio file attachments emailed to a specific address (optional)
* Caller ID The Caller ID number displayed for intercom call
* Analog Phone To assign the analog phone to connect with which FXS port
* Dial Plan The dial restriction, conditions set for making calls

There are also several advanced extension options available. The advanced options establish the connections from the listed extension to other systems within the IPPBX system server. These advanced options include the following:

* Voicemail — The extension support voicemail or not.
* Can reinvite — Can send call again In SIP protocol
* SIP — The extension support SIP protocol
* IAX — The extension support IAX protocol
* NAT — Address reflection between the WAN and public network. When extension is within internal network, it should use NAT
* Call Waiting — The extension support Call Waiting function
* 3-Way Calling — The extension support 3-Way Calling functions
* Codecs — Click here, you can set the extension's codec (voice decode) (default support: alaw, ulaw,g792).

2.3. Trunks

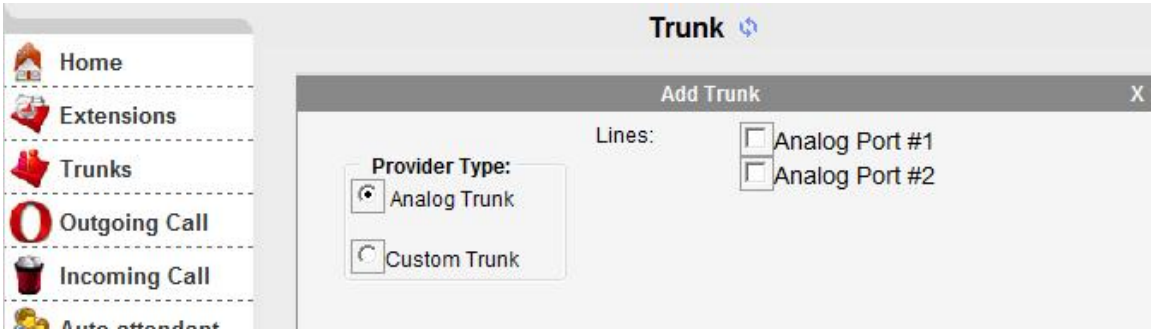
If you want to make external call, you must register with a Trunk in order to connect to the Public Switched Telephone Network (PSTN) or other VoIP service provider. Through the web page you can add a trunk.



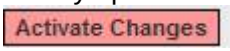
When you click Add a Trunk

, There are two Trunk categories: Analog Trunk, Custom Trunk.

* Analog Trunk — You can enable analog PSTN trunk by this option (FXO port). You need to delete the default Ports 1~6, and Add a PSTN Trunk based on how many FXO port you have actually.



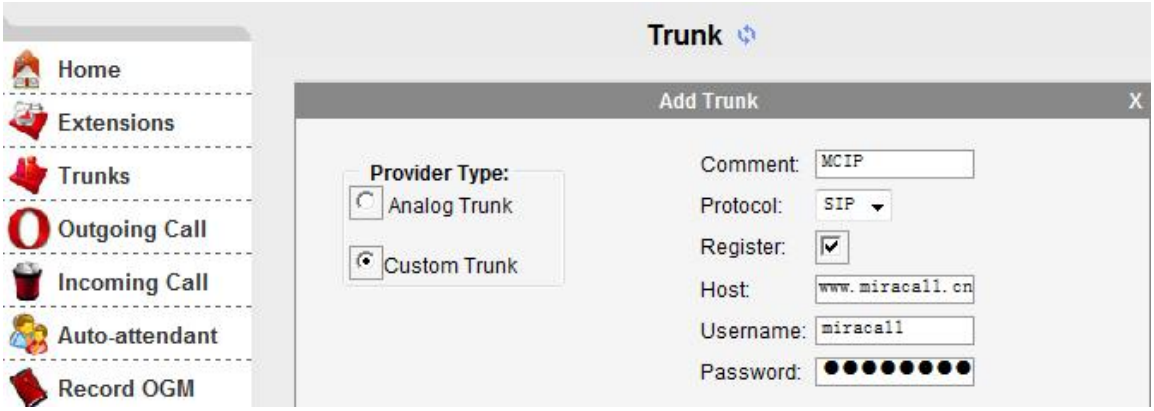
The above map shows the IP PBX has only 2ports Analog trunk. After choose Port 1 and 2, you need to click "Save"

, then 

to enable the setting.

* Custom VoIP Trunk — The Custom VoIP option allows you to create a custom VoIP definition. To create the custom VoIP provider definition you will

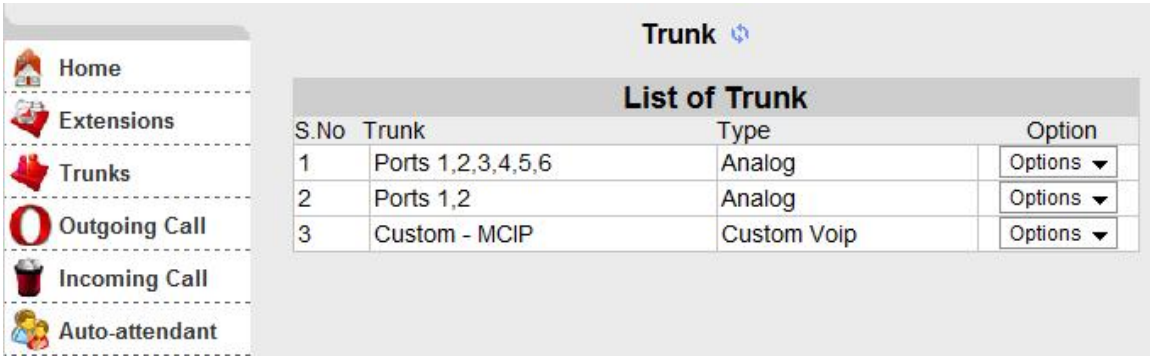
need to complete the following:



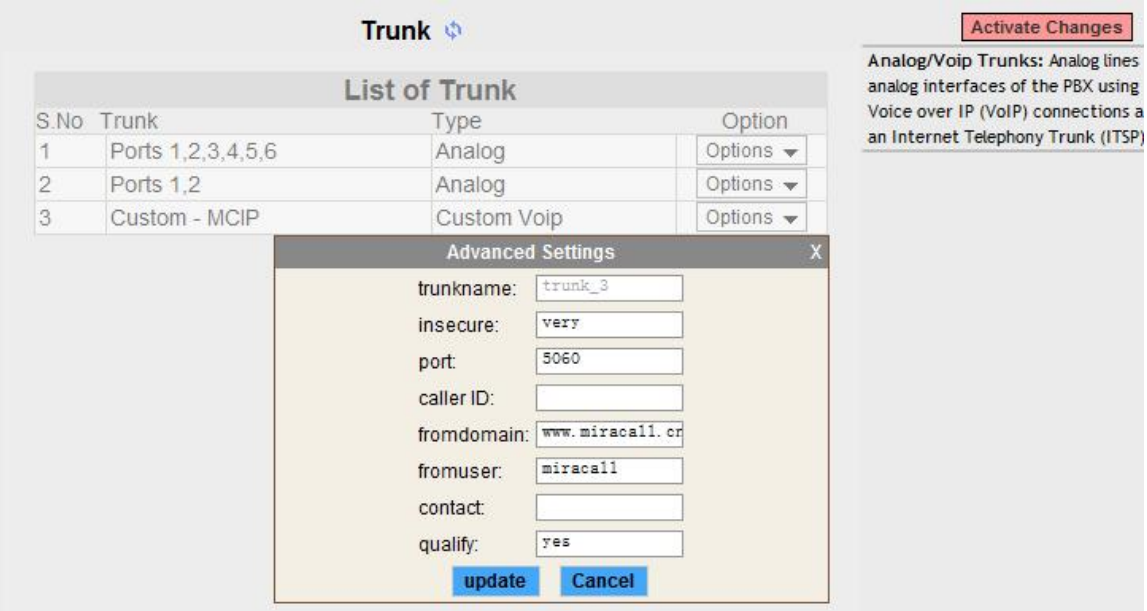
* Comment The name of the SIP Trunk

* Protocol You can select a SIP or IAX protocol
* Register Enable/Disable register.
* Host The IP address provided by your IP carrier provider
* Username The user name provided by your IP carrier provider
* Password The password provided by your IP carrier provider

Once you have added a VoIP Trunk it will appear on the list of Trunk on the Trunk page. There is an Options drop-down list associated with each Trunk listing. The Options drop-down list allows you to edit or delete the Trunk definition, as well as further refine the definition by choosing several advance options. Select either Codecs or Advanced to further refine the definition.



* Codecs Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the internet.
* Advanced The following advanced options are available to further refine your trunk.

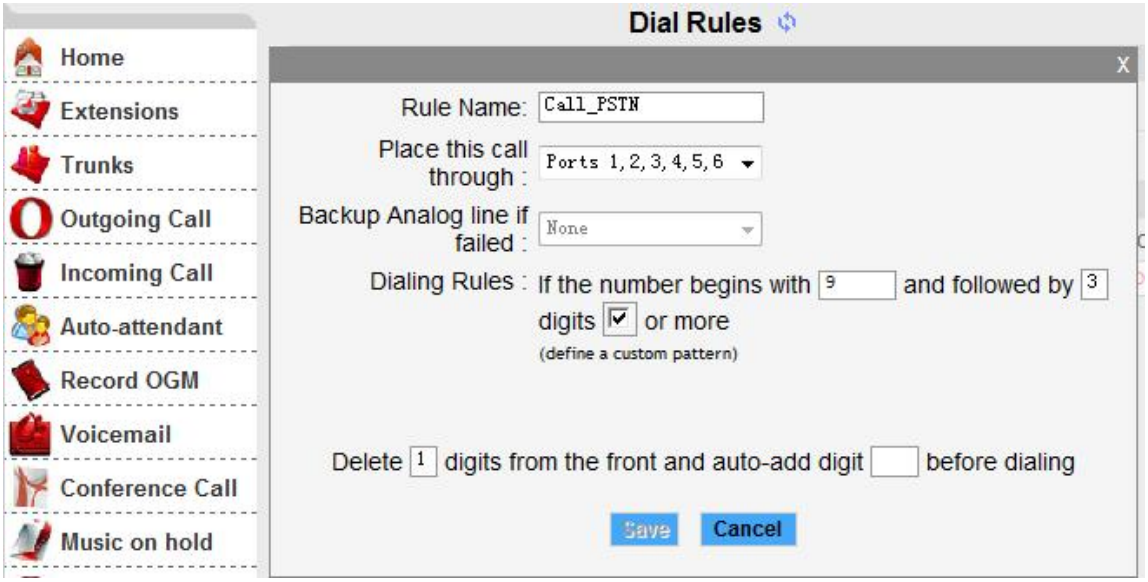


* Trunkname Specify a trunk name if you want to refer to the service provider definition as something other than specified in Comment
* Insecure This option specifies how connections to a service provider (host) should be handled. Valid options are very/yes/no/invite/port. (Default is very)
* Port The register request is sent through the port. (Default is SIP:5060,IAX:4569)
* Caller ID The caller ID will be set to the value specified in this field
* From domain Sets default from: domain in SIP messages when acting as a SIP client.
* From user Sets default from: user in SIP messages when acting as a SIP client
* Contact Specifies a primary extension for call routing

2.4. Outgoing Call (Dial Rules)

The Dial Rules tab on the left toolbar allows you to use basic pattern matching to differentiate outgoing calls and route them accordingly (create different DialPlan).

Click on Add a Dial Rule to define a new DialPlan. The following dialog will be displayed.



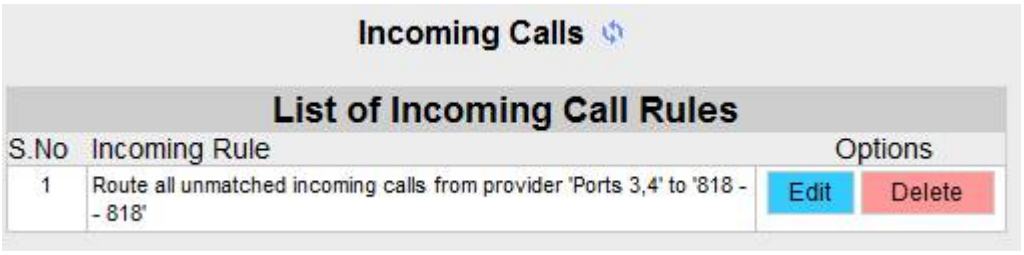
A DialPlan is comprised of the following items:

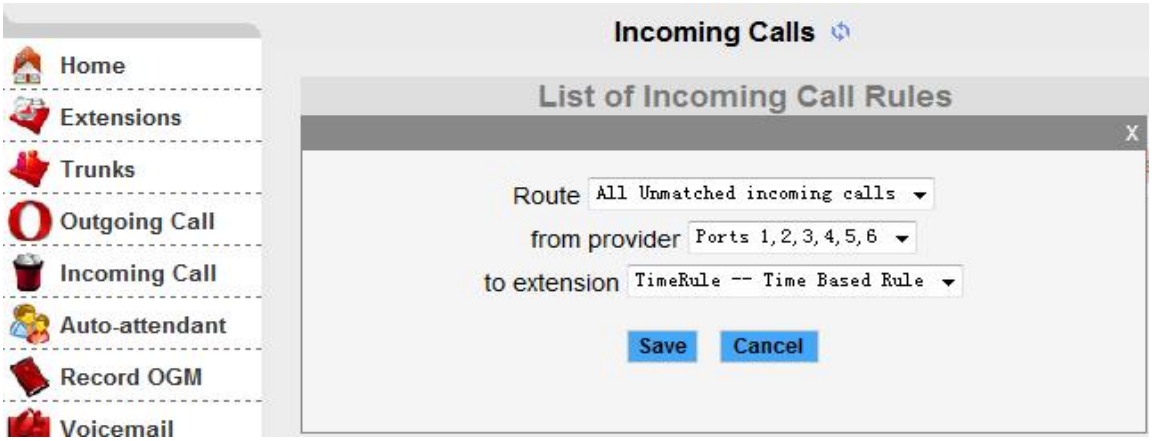
* Rule Name — Set a rule name (can be any)
* Place this call through — Select a Trunk through which the call should be made
* Analog fallback — Select a backup Analog trunk when 1st trunk failed.
* Dialing Rules — The Dialing Rule gives you the ability to use basic pattern matching to differentiate calls and route them accordingly. For instance, if a number begins with 9256, and is followed by 7 or more digits, that would define a call within the state of Alabama. If a call began with 9 followed by 7 digits, it would be a local call

that probably didn't require a long distance charge. Instead of adding a rule for every extension or phone number you call, specify the pattern in this rule similar to the example.

* Strip This option gives you the opportunity to remove specified digits from the call being dialed and replace them with the digits needed to make the call. (same like Dial 9 before making any outside call, like you dial 9+8008108000, in fact, system will strip "9" and the actual number you dial is 8009108000). You can also prepend digits to the beginning of this number (automatic add a digit, such as add 0 before national distance call). Each time after setting ,you should click save Save and Activate Activate Changes .

2.5. Incoming Calls (Incoming Rules)





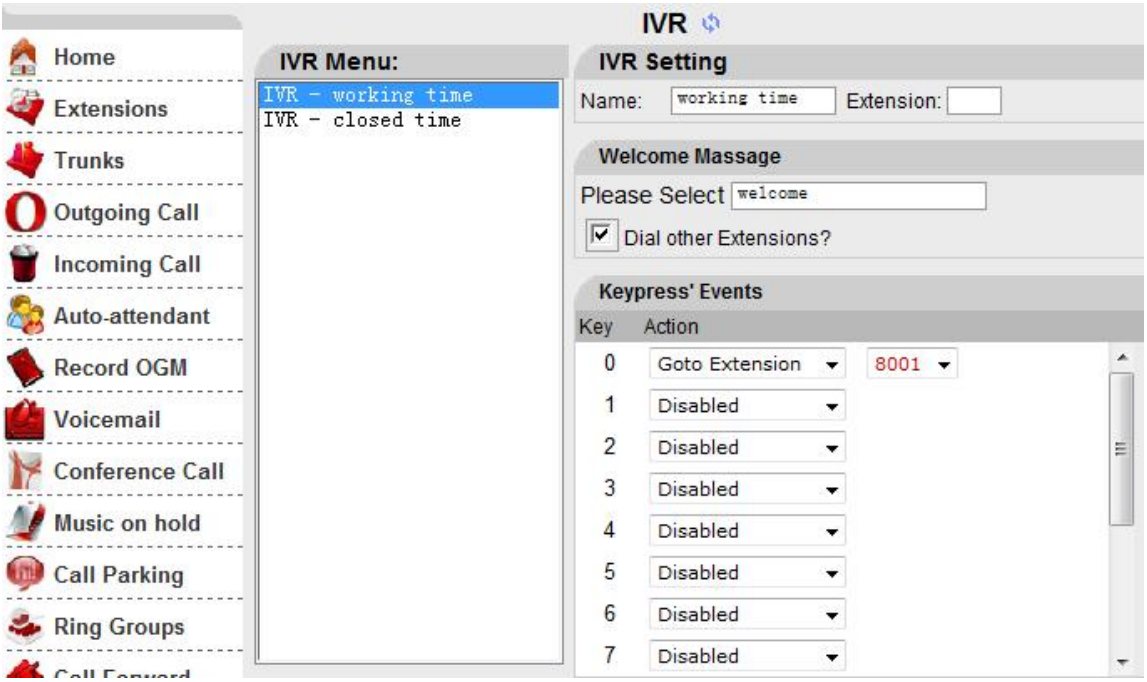
There are only a few options you need to configure

* Route — All Unmatched incoming calls: any call
  Incoming calls that match: the incoming numbers meet some rules, such as some spam numbers, numbers starting from 135xxx ect
* From Provider — From PSTN port or SIP trunk etc.
* To Extension

8001- -User 8001	Means this incoming call will route to extension 8001
8002- -User 8002	Means this incoming call will route to extension 8002 (can be fax)
8028- -User 8028	Means this incoming call will route to extension 8028
8030- -User 8030	Means this incoming call will route to extension 8030
600- voicemail	Means this incoming call will route to voicemail recording.
900--Conference	Means this incoming call will route to conference group
Workingtime- IVR	This incoming call route to OGM of working time
Time-rule -- Time	Means this incoming call will route to time setting rules.
Closed Time- IVR	This incoming call route to OGM of closed time

2.6. Auto-attendant Setting (IVR)

Through the web page, you can create Interactive Voice Response (IVR). IVR are designed to allow for more efficient routing of calls from incoming callers.



Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers.

* Name Set a IVR (OGM) name
* Extension Set an extension which can use this IVR (OGM)
* Please Select Select a IVR message, including the custom message.
* Dial other Extensions? Enable or dis-able to dial other extensions.
* Keypress' Event - Dial any single Key on the phone, then the call will route to…

Dial	Selectable Routes	Routes destination
0	Go to Extension	8001~8032
1	Go to Menu	OGM IVR of Working time, Close time
2	Go to Ring Group	1~ x (you can custom Ringing Groups)
3	Custom	You can define the routes as you like
4	Hangup	Goes on-hook or terminate the conversation
5	Play Invalid	Indicate invalid
6	Disabled	No any action
*	Optional	
#	Optional	
t	Optional	

2.7. Record OGM

In the event that one wants to record custom menu prompts for the IP PBX, which can be used in a IVR, the Record may be used.

A list of previously recorded menus is displayed. Here, the user may modify several options

* Record Again To record a new voice message by phones
* Play To play the message in an extension phone

Delete Delete the recorded voice message

closed.gsm The welcome message to announce company in closed time.

welcome.gms The welcome message to announce company in working time.

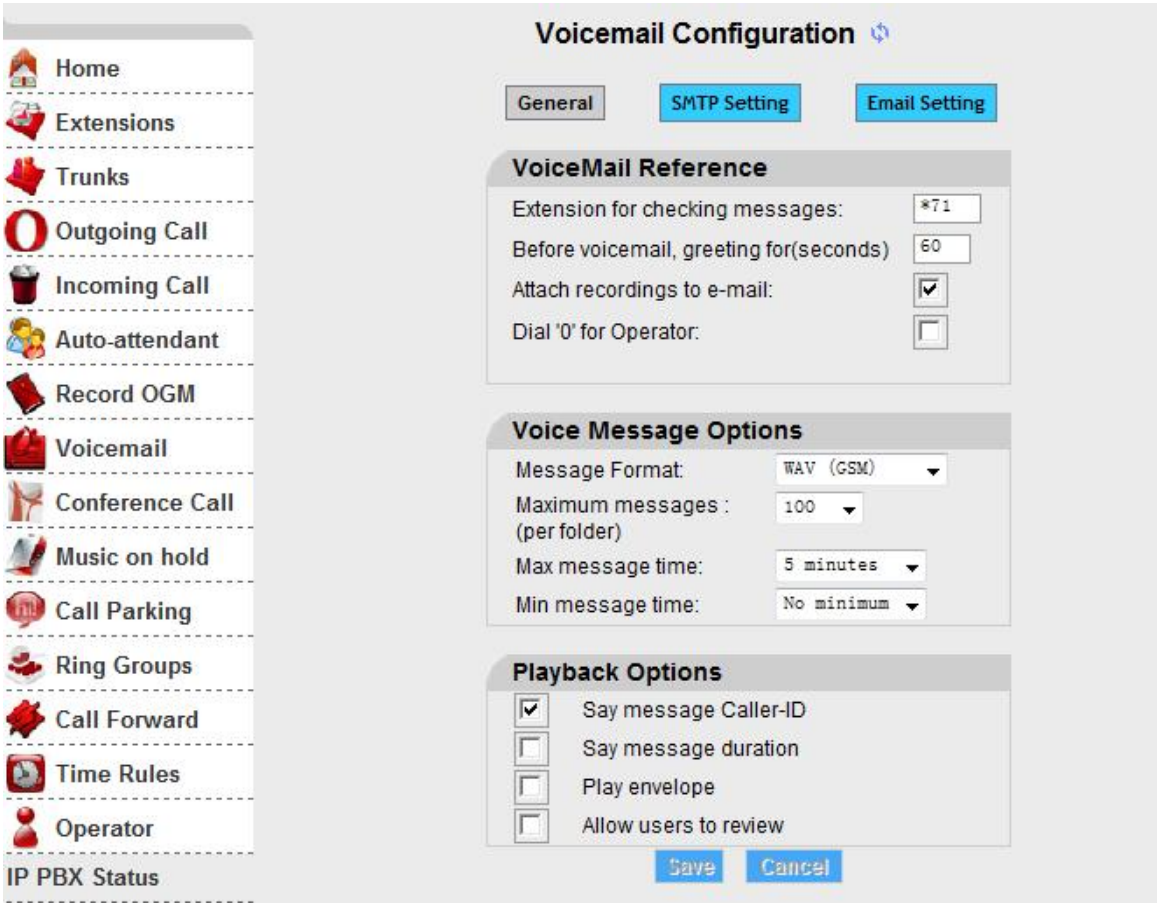
new.gsm The same as welcome.gms

There are two options under “Record a new voice”

* File Name To give a name of the message.
* Extension Used for Recording To select use which extension to record the message.

2.8. Voicemail

Voicemail is an option available for every extension in the IP PBX system. The relationship between the extension and the voice mail is established in the User Extension section of the GUI. You can configure the voicemail through this page.



VoiceMail Reference

The Voicemail Configuration page displays all the extensions to the left, including voice mail. Standard configuration information is also present, allowing you to confirm the extension used to check messages as well as general parameters such as the following:

* Extension for Checking Messages — Dial this number to access personal extension Voicemail box. Need Name and Password.
* Before voicemail, greeting for — The welcome message before recording voicemail . The incoming caller will hear this self-recorded message before leave their message.
* Attach greetings for email — Email the voicemail message to the address you set
* Dial “0” for Operator — Incoming callers press “0” to cancel leaving message in voicemail and switch call to the operator extension.

There are several options that can be specified to define the voicemail message in the system.

Voice Message Options

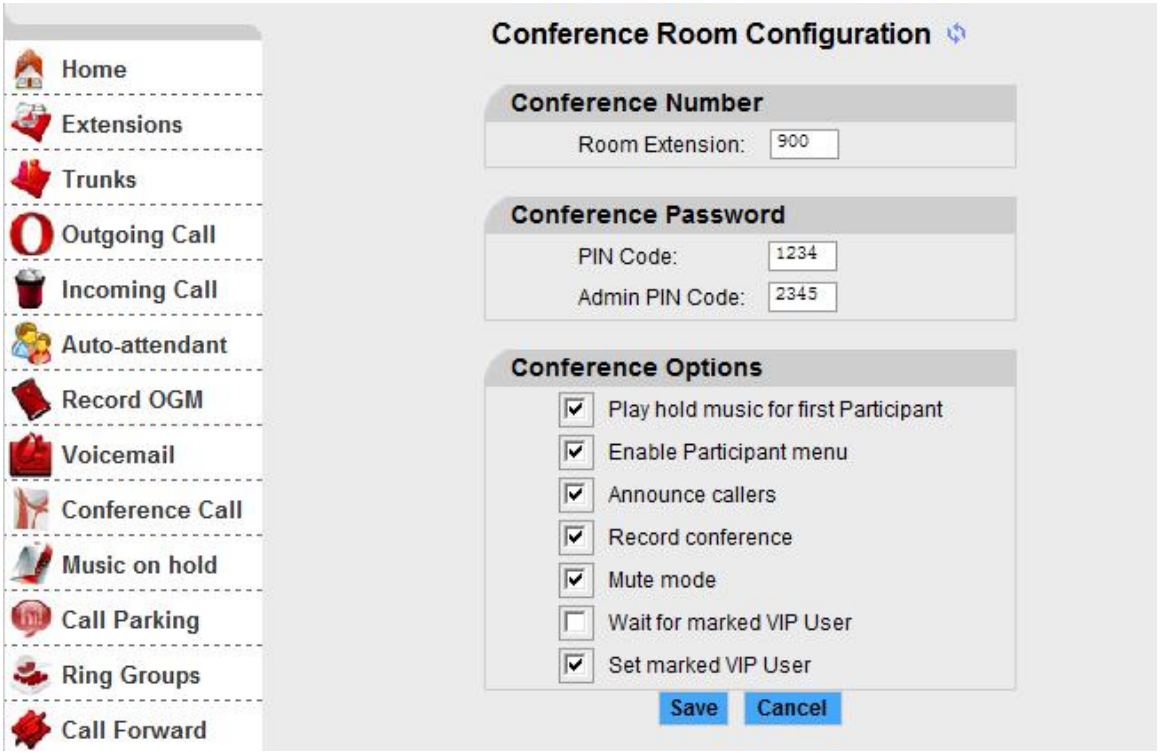
* Message Format	Including WAV (GSM), WAV(16-bits),Raw GSM The format of the voicemail which saved in the IP-PBX
* Maximum Messages	Set the max. messages to leave for each extension user.
* Maximum Message Time	The maximum duration of each message .
* Minimum Message Time	The minimum duration of each message .

Playback Options

* Say Message Caller-ID	It will report the Caller's number before message played
* Say Message Duration	It will report the voice mail duration before message played
* Play Envelope	Report the Date, Time, and Caller's number before the voice mail message played.
* Allow Users to Review	This option provides incoming callers to review their message before saved and can be played back by the owner of the voice mail extension. Standard options are presented to you, allowing you to discard the message or re-record it if you aren't happy with it.

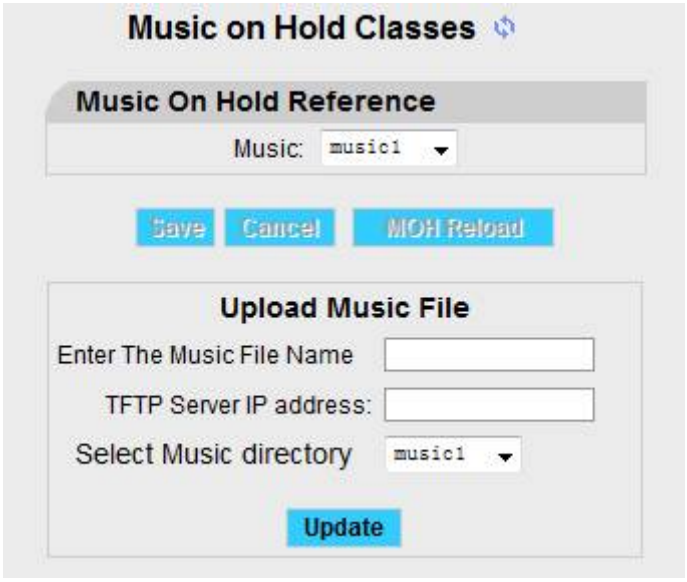
2.9. Conference Call

Allow more extension people to make a conference call meeting.



Action	Explain	Default
Room Extension	Dial this extension to entry to conference group.	900
Pin Code	Input this password before login in conference	1234
Admin Pin Code	Administer can use this code to manage the conference room.	1234
Conference Optional		
Play hold music for first participant	The first joiner hear the waiting music	
Enable participant Menu	The setting menu for conference parties extension	
Announce Callers	Make announcement for the joiners	
Record Conference	Record the conference meeting .	
Mute Mode	No voice mode	
Wait for Marked VIP user	The important user is going to join the conference	
Set Marked VIP user	To set the important user .	

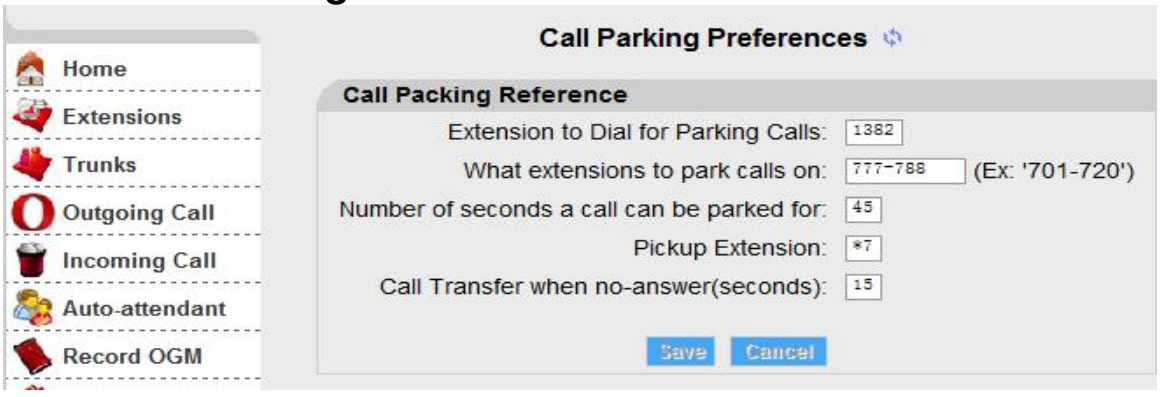
2.10. Music On Hold



Music: In default setting,there are music 1~10 .

Upload Music File : to upload a holding music from TFTP Server.

2.11. Call Parking

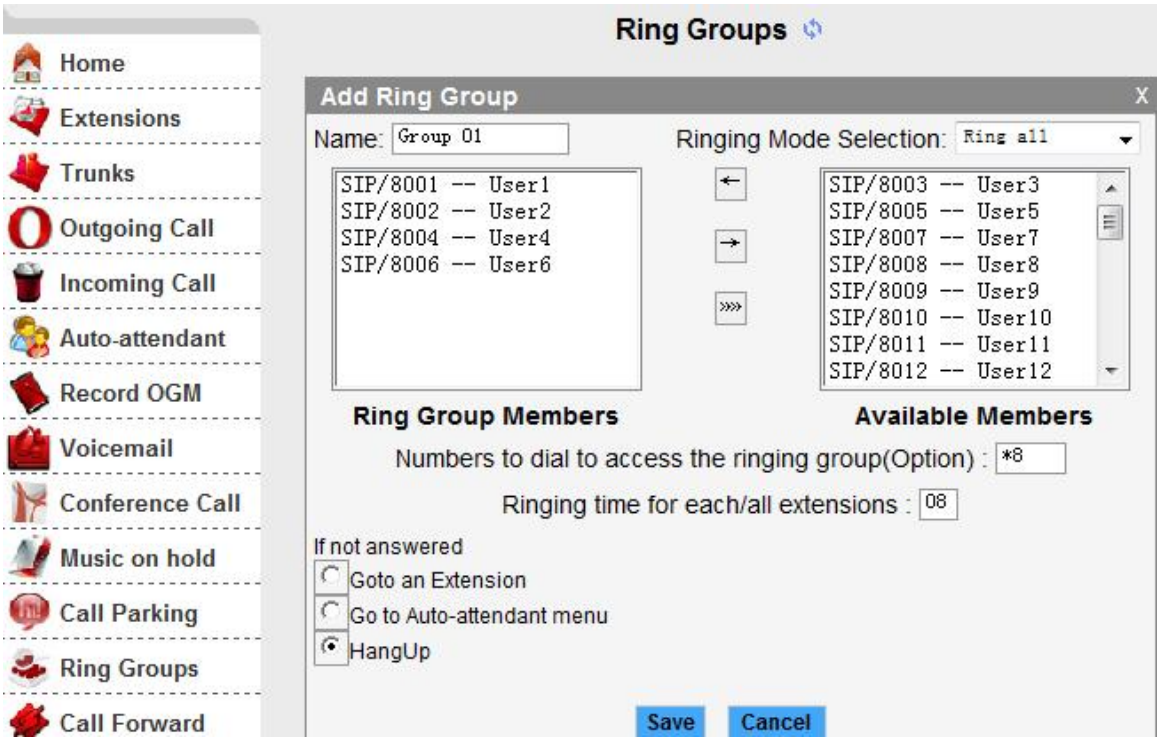


Call park allow you to place the incoming caller on hold. Any extension can talk to this parked caller. The parked party will hear the music.

Extension to Dial for Parking Calls:	Set a Call Parking number. To park the call, the incoming Caller should be transfer to this Parked number.
What extensions to park call on:	Set the number dialed for call park (eg: 701-720) , each time for Park, system will assign a number automatically for the Parked party, dial this number to retrieve the conversations.
Number of seconds a call can be parked for:	Set duration the Parked party can be parked for. It will automatically return to the last conversation extension when the parked time finished.
Pickup Extension:	Set the code to pick up ringing extension.
Call transfer when no-ansewr:	Set the duration over time on no-answer.

2.12. Ring Groups

The extensions can be a group . Each extension will ring within the same group for the incoming call. Each ringing extension can pick up this call.



Define Ring Groups to Dial more than one extension

* Name Set a Ring Group name, can not input special characters.
* Strategy There is a drop-down list, you can choose Ring all or Ring in order. Ring all: ringing to all extensions until call answered . Ring in order: ring one by one

* Ring Group Members Add Ring Group member from available right side.

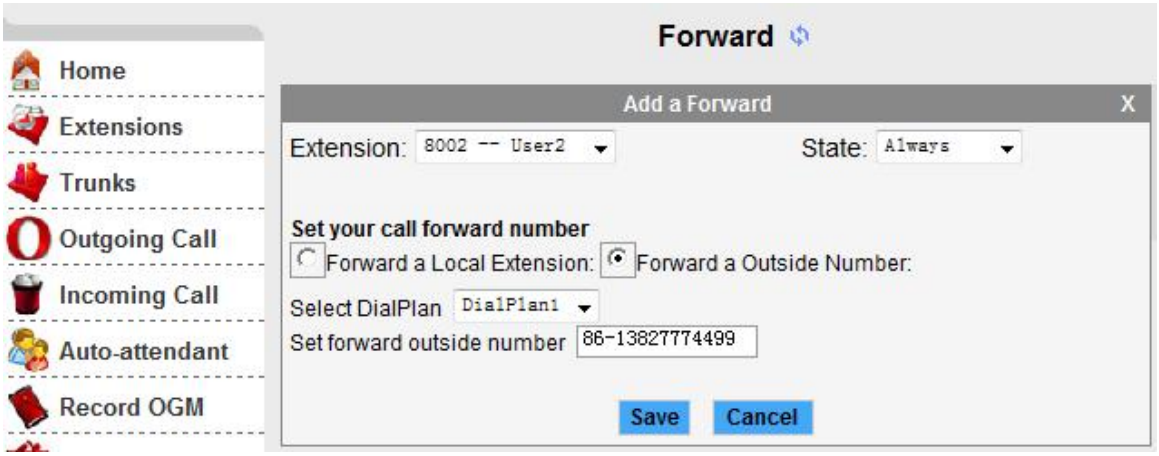
When there's no answer ,you can choose to route the call to Voicemail, IVR or hand-up.

* Number to dial to access ringing group — The access code to ringing group.

* Ring time for each/all extension — The duration ringing for each extension in the group, after duration finished , call will ring to another circle or another extension.

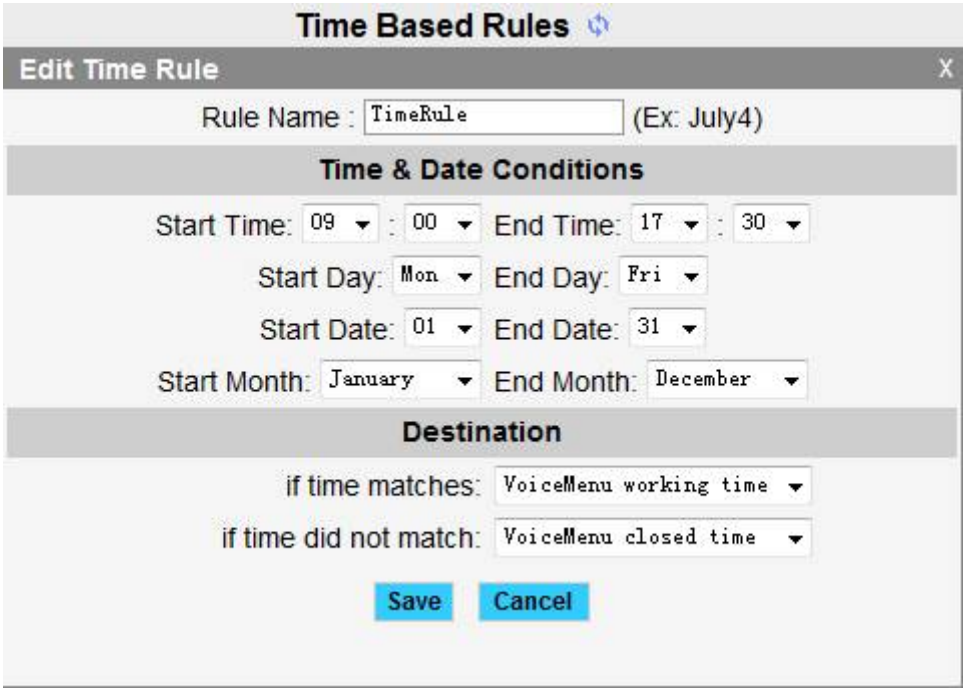
2.13. Call Forward

This rules allow the extension forward to intercom or external line/number, Immediately or On Busy, No-one Answer the call



2.14. Time Rules

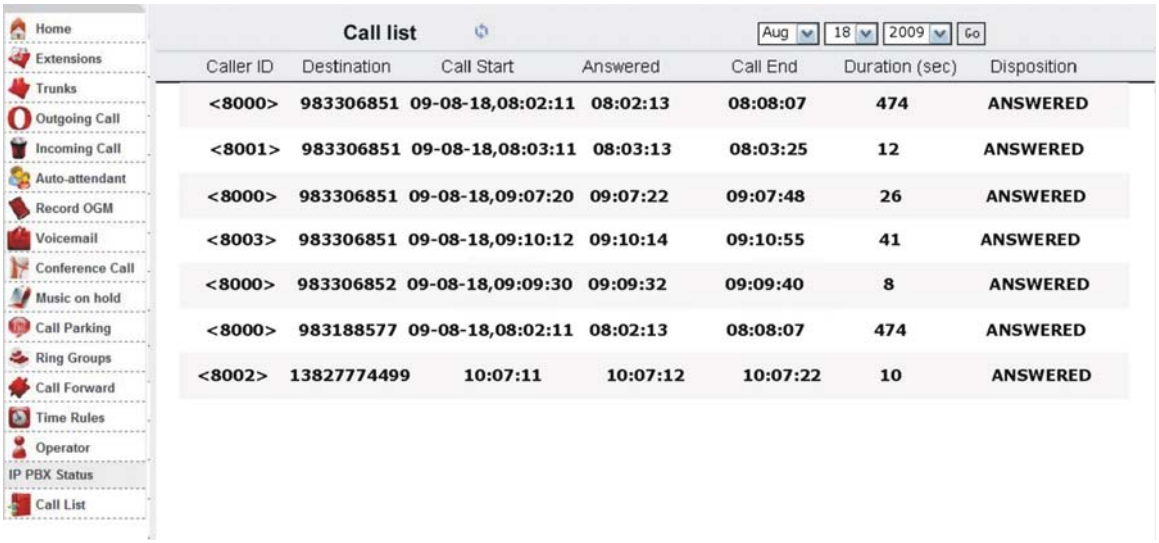
To set the times and date of the system. You can select the route destination when the time is matched or not.



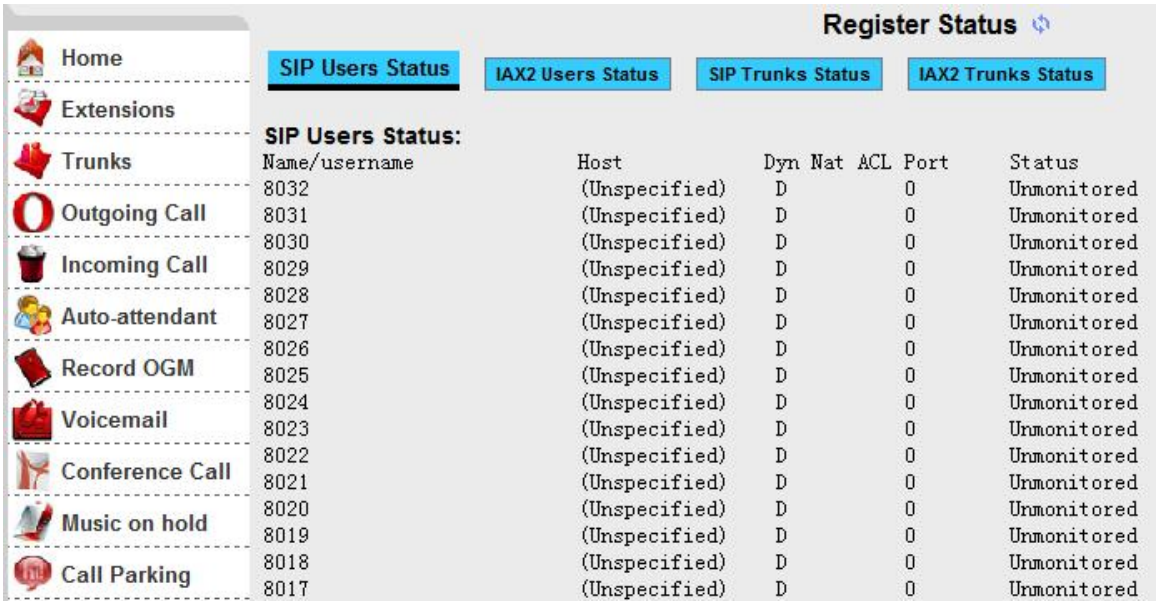
Chapter 3 IP-PBX Status

3.1. Call List

To show calling history, starting time, duration, answered or un-answered status .

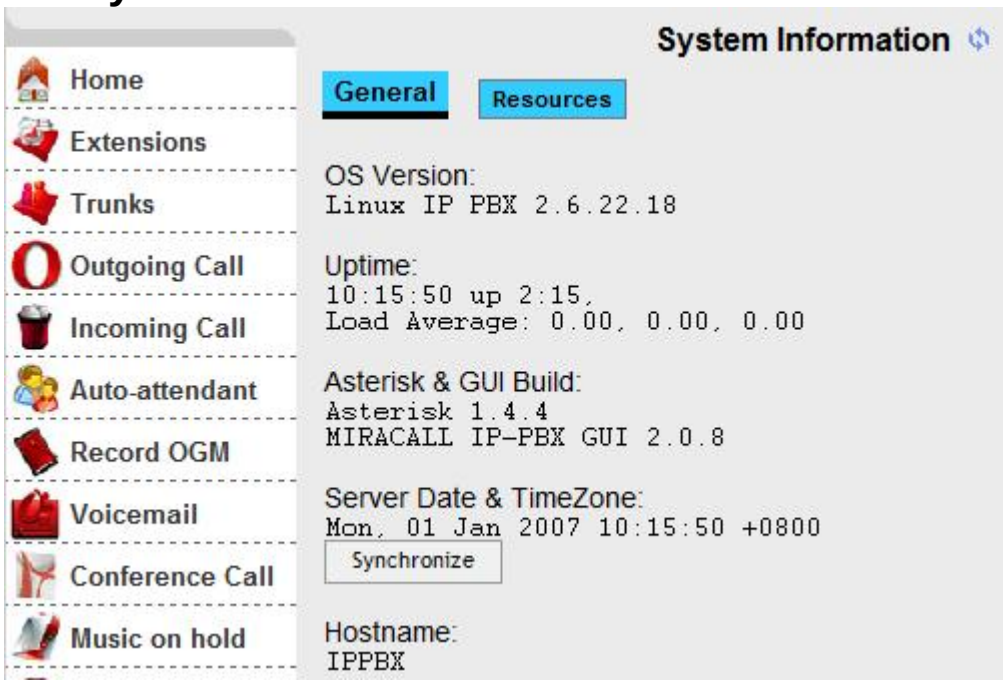


3.2. Register Status



If there's IP address displayed then it mean the extension is registered successfully.

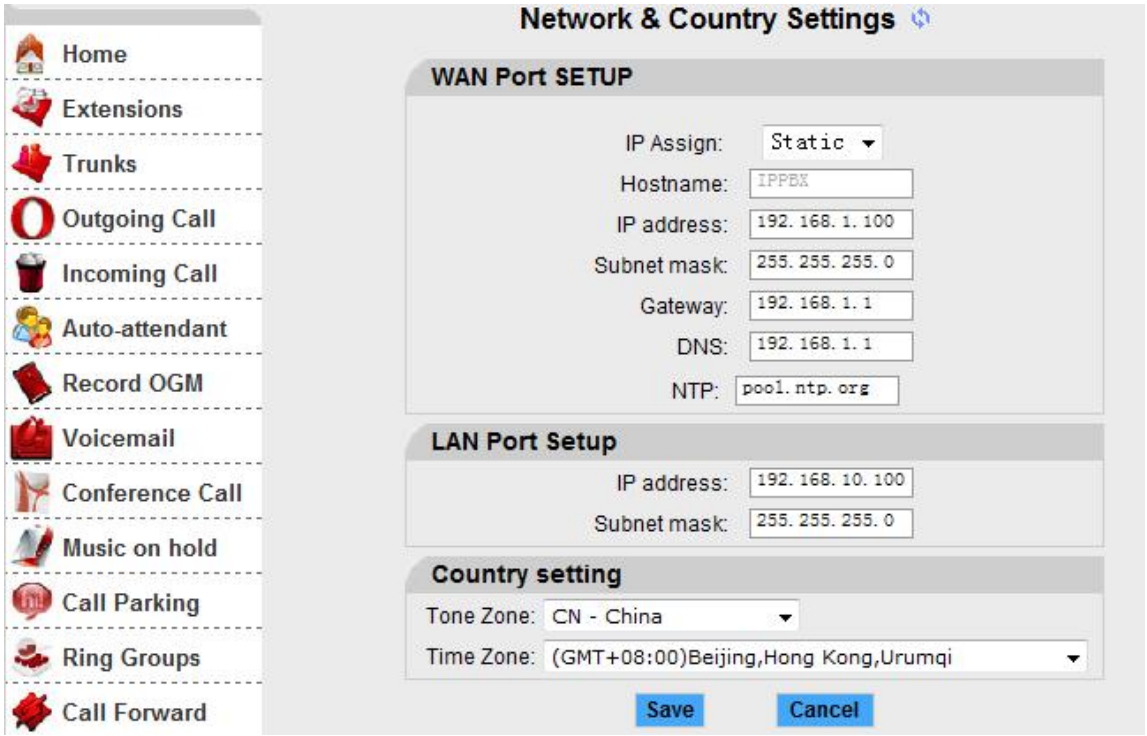
3.3. System Info



To display the IP-PBX system data and information , version, Asterisk version etc.

Chapter4 System Management

4.1. Network and Country



On this page you can set WAN, LAN interface information and country.

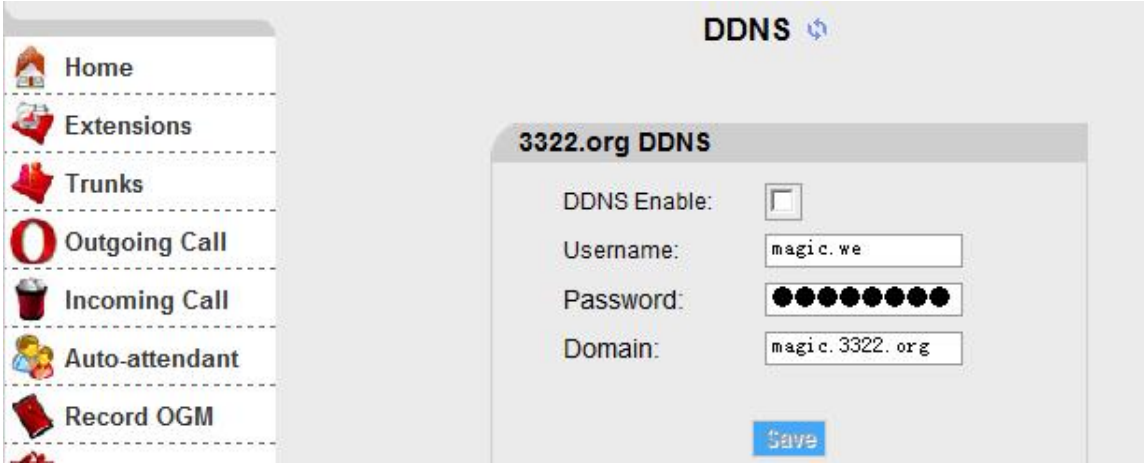
IP Assign: you can select STATIC, DHCP and PPPoE three mode

NTP: Set NTP server address.

Country: Set your Country, and use the Country Zone

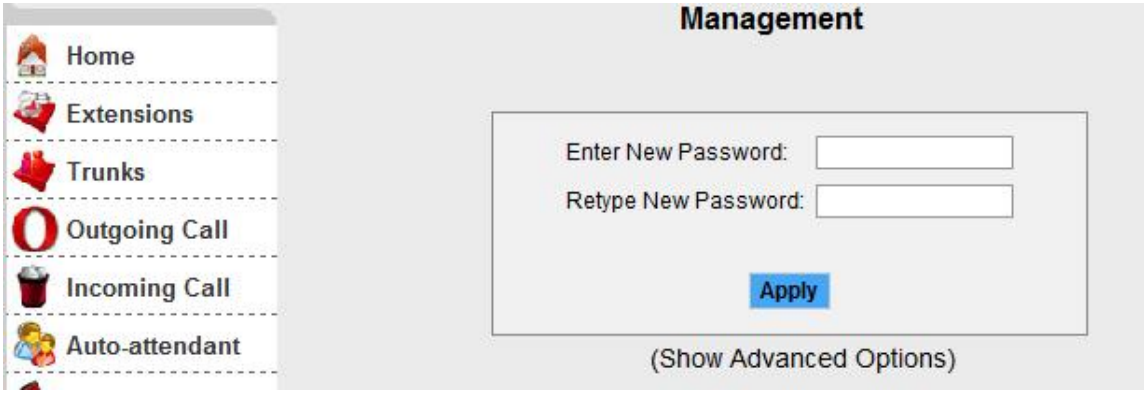
4.2. DDNS

On this page, you can set DDNS reference.



Notice: Now, it only support 3322.org server. More other servers, you can customize your requirement

4.3. Change Password



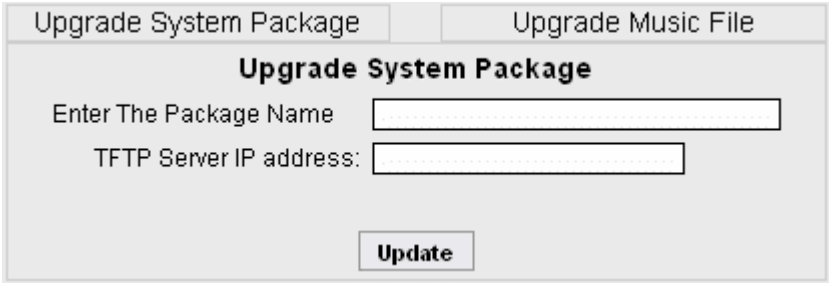
On this page, you can change the administrator password (Default password: admin)

4.4. Backup

On this page, clicking the “Take a Backup” button, you can backup nonce configuration

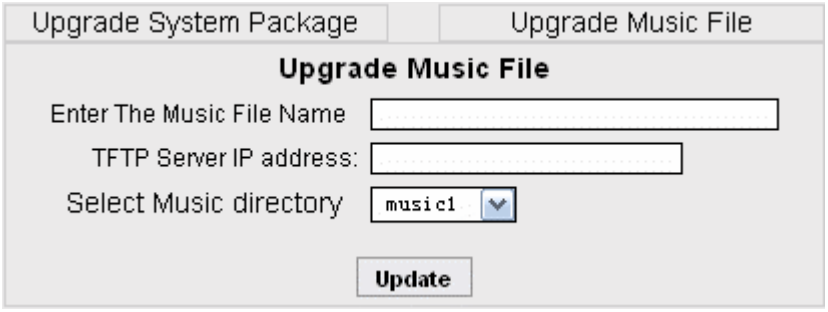
4.5. Upgrade

In this page you can upgrade system package and music file Upgrade System Package



* Enter The Package Name Set system package name
* TFTP Server IP address Set TFTP server IP

Upgrade Music File



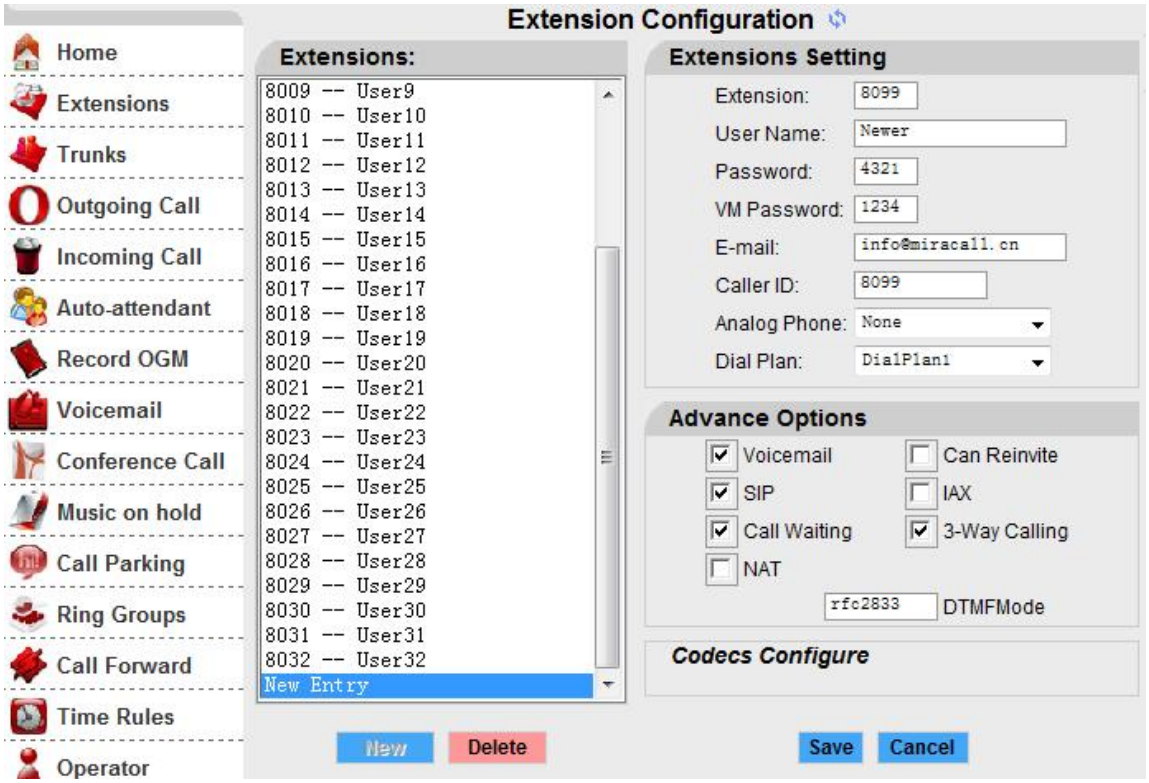
* Enter The Music File Name — Set you want upgrade music file name
* TFTP Server IP address — Set the TFTP server IP
* Select Music directory — Select directory that you want saved music file.

Chapter 5 Operating Instruction

5.1. How to make a internal call

Add a New User

Users-> New:



Set new user Extension is 8099….

Name, Password and Caller ID, etc……

Select Dial Plan is DialPlan1

Use a SIP/IAX IP Phone registered with the user.
Then you can use 801 call 802 successful.

5.2. How to make a outbound call

To make an outbound call, we need to add a trunk first. There are two types of Trunk:

Analog Ports: FXO ports of MC-IP8008, connect to local PSTN
VoIP Trunk: SIP or IAX trunk, connect to remote SIP/IAX server

In the MC-IP8008 I am using, the port1-4 are configured as FXO ports, port5-8 are configured as FXS ports. When a port is configured as FXO port, the corresponding LED shows RED. When a port is configured as FXS port, the corresponding LED shows GREEN.

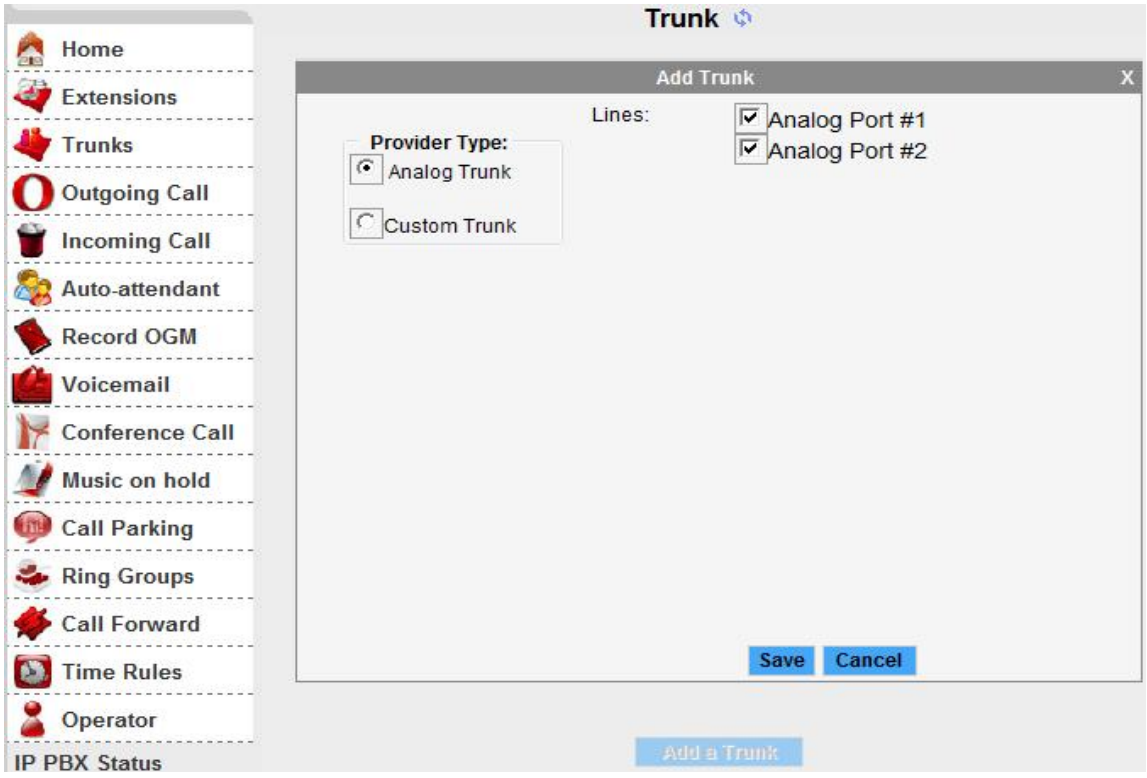
What are FXO and FXS?

FXS (Foreign eXchange Station) is an interface which drives a telephone or FAX machine. FXS interfaces get phones plugged into them, delivery battery, and provide ringing. FXS interfaces are signalled with FXO signalling. **FXO** (Foreign eXchange Office) is an interface that connects to a phone line. They supply your PBX with access to the public telephone network. FXO interfaces use FXS signalling. FXS interfaces are that allow you to hook telephones to your PBX, and FXO interfaces allow you to connect your PBX to real analog phone lines.

Make call via PSTN trunk

Add Analog Trunk

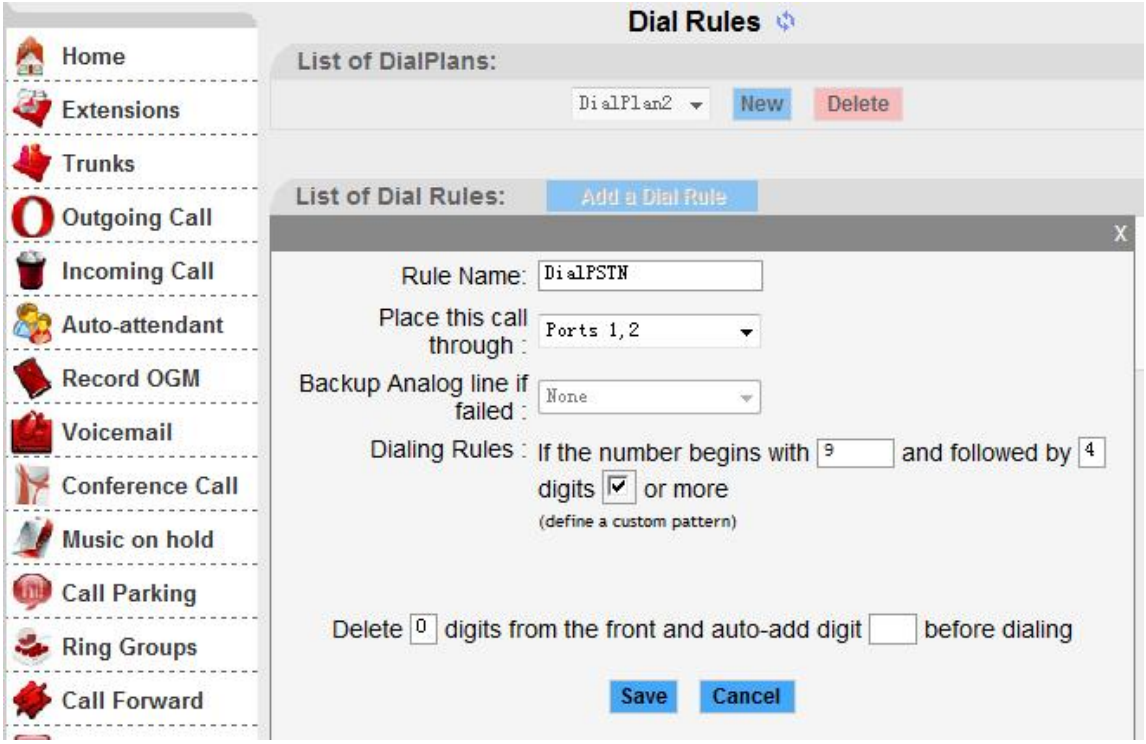
Trunk -> Add a Trunk:



Add Dial Rule

In a Dial Rule -> add a Dial rule as below

Dial Rules



We have now added a Dial rule “Dial_PSTN” in the “DialPlan2”.

As we can see from the dialing rule of “Dial_PSTN”, all numbers start with 9 will be cut the first digit (‘9’) .

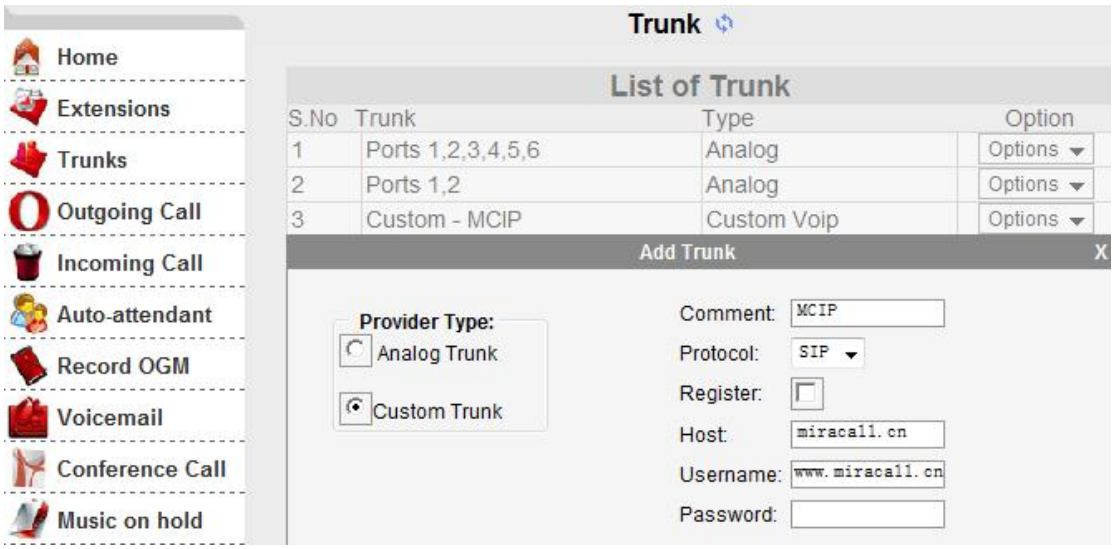
After we have done above, in the extension we can dial 9 + local number to dial out via SIP line.

Make call via VoIP trunk

Add VoIP service provider

Trunk -> Add a Trunk:

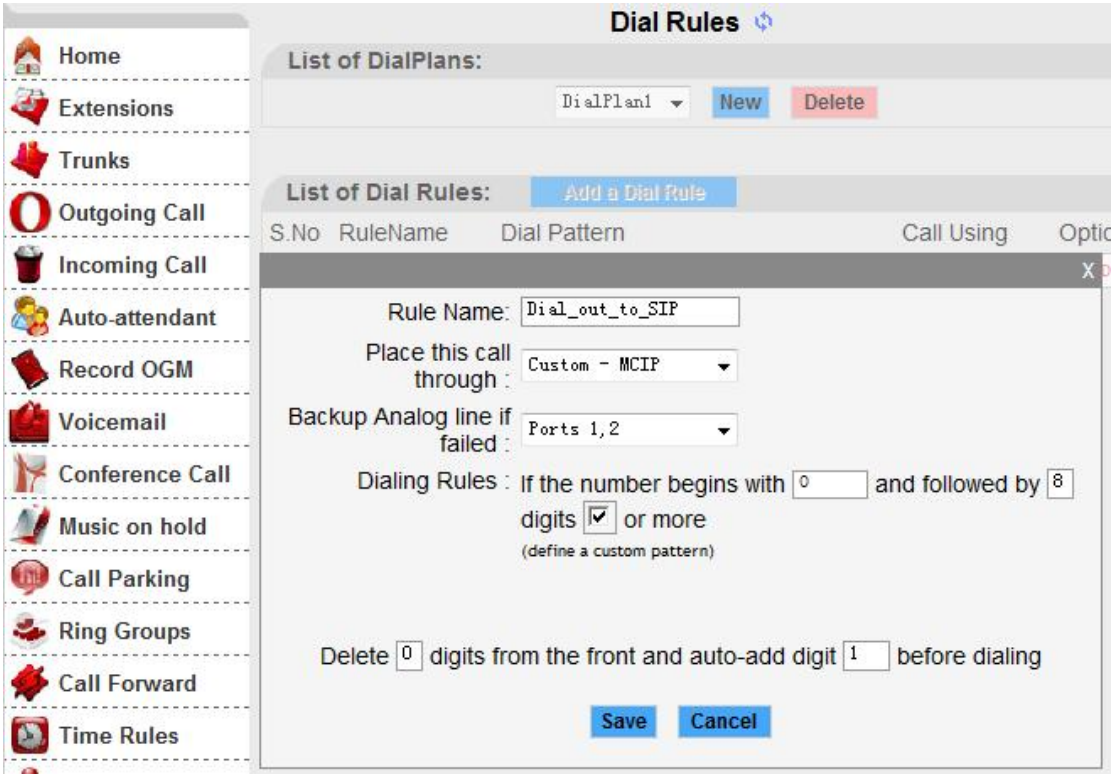
Add a Custom Trunk



Add Dial Rule

In Dial Rules -> add a new calling rule as below

Dial Rules



Now we have added a new calling rule “Dail_Out_to_SIP” in the “DialPlan1”.

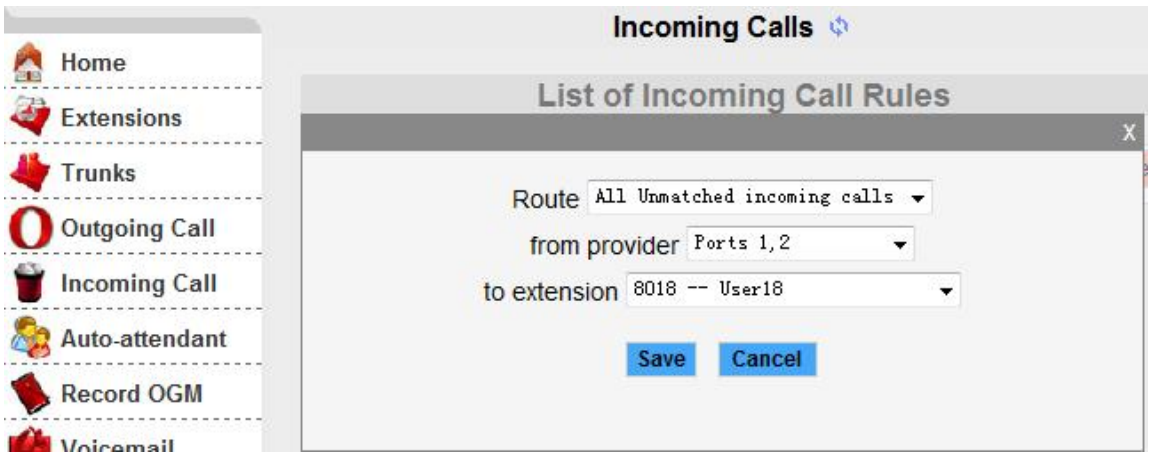
As we can see from the dialing rule of “Out_VoIP”, all numbers start with 0 will be cut the first one digits (‘0’) and sent to my sip service provide voip.

The Out_VoIP is in the same DialPlan1. Since we have added this dial plan to the extensions in above, we don’t need to add dial plan again.

So far when we have added two calling rules, any call start with 9 will be route to PSTN, and call starts with 0 will be route to VoIP.

5.3. How to make an incoming call

Add an Incoming call. Click “add a incoming call rule”



Select Route “All Unmatched incoming calls” From provider “Port 1,2”

To extension “8018 – IP Phone” (here, you can select a extension, a IVR or others) Then, if there is incoming call from Port1,2 channel the extension 8018 will ring